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1/77

19JUN03 E816272-3 D01063
P01/7700 0.00-0314252.8

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1.	Your reference	RL.P52773GB 19 JUN 2003
2.	Patent application number *(The Patent Office will fill in this part)*	0314252.8
3.	Full name, address and postcode of the or of each applicant *(underline all surnames)*	Telefonaktiebolaget LM Ericsson (Publ) SE-12625 Stockholm Sweden
	Patents ADP number *(if you know it)*	6069744001
	If the applicant is a corporate body, give the country/state of its incorporation	Sweden
4.	Title of the invention	Conversational Bearer Negotiation
5.	Name of your agent *(if you have one)*	Marks & Clerk
	"Address for service" in the United Kingdom to which all correspondence should be sent *(including the postcode)*	4220 Nash Court Oxford Business Park South Oxford OX4 2RU United Kingdom
	Patents ADP number *(if you know it)*	7271125001
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7.	If this application is divided or otherwise derived from an earlier UK application, give the number and the filing date of the earlier application	Number of earlier application / Date of filing *(day / month / year)*
8.	Is a statement of inventorship and of right to grant of a patent required in support of this request? *(Answer 'Yes' if:* *a) any applicant named in part 3 is not an inventor, or* *b) there is an inventor who is not named as an applicant, or* *c) any named applicant is a corporate body.* *See note (d))*	Yes

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Continuation sheets of this form		
Description	10	
Claim *(s)*	1	CP
Abstract	1	
Drawing *(s)*	5 +5	

10. If you are also filing any of the following, state how many against each item.

Priority documents

Translations of priority documents

Statement of inventorship and right to grant of a patent *(Patents Form 7/77)*

Request for preliminary examination and search *(Patents Form 9/77)*

Request for substantive examination *(Patents Form 10/77)*

Any other documents *(please specify)*

11. I/We request the grant of a patent on the basis of this application.

Signature Marks & Clerk Date

Marks & Clerk 18 June 2003

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Dr. Robert Lind
01865-397900

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CONVERSATIONAL BEARER NEGOTIATION

Field of the Invention

The present invention relates to negotiating the setting up of conversational bearers in communication networks, which bearers can be used, for example, to carry real time voice and video information.

Background to the Invention

Digital cellular telephone networks have traditionally relied upon circuit switched channels to carry user traffic such as voice communications. A circuit switched channel is formed by the allocation of one slot per frame in a given TDMA channel. Whilst circuit switched sessions have proved adequate for voice calls, they do not provide an efficient mechanism for transferring large amounts of data which is "bursty" in nature. For example, the setting up of a circuit switched session to download a web page from a web server is likely to result in the connection remaining idle for significant amounts of time, and being overloaded when there is data to transmit.

To facilitate fast data transfers to mobile terminals, packet switched data services are being introduced to digital cellular telephone networks. For example, the General Packet Radio Service (GPRS) is currently being introduced to many GSM networks. Unlike circuit switched sessions, a GPRS session (referred to as a PDP context) for a given user does not necessarily occupy one slot per frame on a given TDMA channel. Rather, slots are only used when the user has data to send or receive. When there is no traffic to transmit, no slots are allocated to the user. When there is a large volume of data to transmit, the user may be allocated one or more slots per frame.

GPRS will be available in future third generation networks such as 3G networks which will rely upon CDMA rather than TDMA. 3G networks will however continue to provide circuit switched services at least for the foreseeable future, although these sessions will not necessarily be end to end. Rather, the links between mobile terminals and the networks will be circuit switched, with data being routed within and between

networks via high capacity packet switched networks (which have sufficient bandwidth to handle real time traffic).

It is envisaged that in the future, the packet switched (access) domain will be able to carry real time information streams, for example relating to voice and video telephony. However, at present the transmission reliability of GPRS is not sufficient to provide users with telephony services of the quality which they will expect, hence the continued provision of circuit switched services (the provision of circuit switched services is also likely to be necessary by the need to continue to service older mobile terminal equipment which is not GPRS enabled).

To facilitate the provision of multimedia services via the packet switched "domain", the 3rd Generation Partnership project (3GPP) responsible for the 3G standards has been developing a so-called IP Multimedia Core Network Subsystem (IMS). IMS communicates with the GPRS core network and contains all elements that are used to provide IP based multimedia services. For a mobile to mobile call, and assuming the mobiles belong to different networks, an IMS will be provided in each mobile's home network. Each IMS is connected to the GPRS core network of its home network. The base protocol for multimedia services is the IETF Session Initiation Protocol (SIP). SIP makes it possible for a calling party to establish a packet switched session to a called party (using so-called SIP User Agents, UAs, installed in the user terminals) even though the calling party does not know the current IP address of the called party prior to initiating the call. SIP provides other functionality including the negotiation of session parameters (e.g. Quality of Service and codecs).

Figure 1 illustrates schematically a 3G network providing circuit switched (CS) and packet switched (PS) access networks to a mobile terminal. The figure illustrates a call being made by the mobile terminal, via its circuit switched access network, to a PC which has access only to a packet switched network. The session is initiated by the dialling of a telephone number from the mobile terminal, i.e. this does not involve any exchange of SIP signalling between the home network and the mobile terminal, and SIP URLs cannot be transferred over the CS domain. The destination terminal must have allocated to it a standard telephone number in order for such a session to be established.

Translation between circuit switched and packet switched data is performed by an interworking gateway (GW), with the GW establishing the packet switched session to the PC using SIP signalling. Assuming that the packet switched network used by the PC has sufficient bandwidth (e.g. the network is a broadband network), the call will provide the users with a sufficient level of quality for voice and video. In this scenario, the IMS of the home operator's network is not used.

In addition to the need for the destination terminal to have allocated to it a telephone number, a further disadvantage of the architecture of Figure 1 is that the destination terminal will not necessarily know that a conversational bearer has been established using a CS access network. Any attempt by the destination terminal to establish some additional (non-conversational) PS bearer will fail, because the gateway cannot provide this service. Also, any attempt by the initiating terminal to establish a (non-conversational) PS bearer may fail because the destination terminal will not be able to associate the set-up request with the existing conversational bearer.

Figure 2 illustrates an alternative scenario in which a call between the mobile terminal and the PC is established using the PS access network available to the mobile terminal. The call is established using a SIP server of the IMS. Due to the limited bandwidth of the PS access network available to the mobile terminal, the session is unlikely to be of sufficient quality to handle real time voice and video data. A separate CS bearer should be established for this purpose. However, this might not be straightforward given that the initiating or terminating terminal might know only the SIP URL of the peer terminal, and not its telephone number.

It is likely that users will prefer to initiate and receive circuit switched and packet switched calls using a common signalling interface. However, under the current proposals, a user would initiate and receive a packet switched call using SIP, e.g. to initiate a packet switched call the user would enter the SIP address for the called party (e.g. john@example.org), whilst he/she would initiate and receive a circuit switched call using the DTAP protocol, e.g. to initiate such a call the user would dial the called party's telephone number (e.g. 012345...). Network operators would also prefer to use a common signalling interface as this will ease the migration of circuit switched

services to the packet switched domain, when that domain has evolved sufficiently to provide the required services.

Figure 1 illustrates a session established between two peer nodes of a telecommunications system over circuit switched and packet switched access networks;
Figure 2 illustrates a session established between two peer nodes of a telecommunications system over respective packet switched access networks;
Figure 3 illustrates in detail an architecture for allowing a packet switched session to be established between peer mobile terminals using SIP; and
Figure 4 illustrates a procedure for setting up a conversational bearer in the CS domain using signalling sent over the PS domain;
Figures 5 and 6 illustrate signalling associated with the setting up of a session, extending at least in part over a circuit switched network, using a packet switched network to carry the set-up signalling.

Typical call session scenarios in existing and proposed telecommunication networks have been described above with reference to Figures 1 and 2.

Figure 3 illustrates schematically a typical scenario where the user of a mobile terminal or "User Equipment" (UE) 1 is a subscriber of a 3G cellular telephone network 2 (the subscriber's home network). The UE 1 is a dual mode terminal, e.g. as specified in 3GPP Release 5 (dual CS-IMS/PS). The subscriber using the UE is identified in the home network 2 by a unique subscriber identity (e.g. International Mobile Subscriber Identity, IMSI), and the network is referred to as the subscriber's "home" network. The home network comprises a General Packet Radio Service (GPRS) core network 3 and a circuit switched core network 4. Both the core networks 3,4 make use of a common UMTS Radio Access Network (UTRAN) 5. In addition to or as an alternative to the UTRAN, a UE may communicate with the core networks via a GERAN (GSM/EDGE Radio Access Network).

Within the GPRS network 3, two nodes relevant to the UE 1 can be identified. These are the Serving GPRS Support node (SGSN) 6 and the Gateway GPRS Support Node (GGSN) 7. The role of the SGSN 6 is to maintain subscription data - identities and

addresses - and to track the location of the UE 1 within the network. The role of the GGSN 7 is to maintain subscription information and allocated IP addresses and to track the SGSN to which the UE 1 is attached. The GGSN 7 is coupled to an IP backbone network 8 (the SGSN is also coupled to the IP network 8, although this session is not shown in the Figure – communication between nodes of the GPRS network, including the GGSN and the SGSN, and between gateway nodes of the UTRAN and the GPRS network, will take place via the IP network 8). Typically, when the UE 1 is turned on it "attaches" itself to the GGSN and a PDP context is established between the UE 1 and the GGSN 7. This context provides a "pipe" for transporting data from the UE 1 to the GGSN 7. This process involves the allocation of an IP address to the UE 1. Typically, the routing prefix part of the address is a routing prefix allocated to the GGSN 7.

Also illustrated in Figure 3 is an IP Multimedia Core Network Subsystem (IMS) 9 which contains all of the elements required to provide IP based multimedia services in the packet switched domain, and which communicates with mobile terminals. The functionality provided by the IMS 9 is set out in 3GPP V5.6.0. The IMS 9 consists of a set of nodes which communicate between themselves and with nodes outside of the IMS via the IP backbone network 8 (these sessions are not shown in the Figure). Illustrated within the IMS 9 are a proxy call state control function (P-CSCF) node 10 and a serving call state control function (S-CSCF) node 11. It is assumed here that the IMS is owned by the operator of the home network 2 (although this need not be the case). In the case of a roaming subscriber, the UTRAN and core networks will of course belong to a "visited" network. The P-CSCF will also belong to the visited network, whilst the S-CSCF and the HSS (Home Subscriber Server) will be located in the home network. A subscriber is identified within the IMS by an IMPI (IP multimedia private identity) which has a unique relation with the IMS subscription.

The S-CSCF 11 performs the session control services for the UE, and maintains a session state as needed by the home network operator for support of services. The main function performed by the S-CSCF 11 during a session is the routing of incoming and outgoing call set-up requests. The main function performed by the P-CSCF 10 is to route SIP messages between the UE 1 and the IMS 9 of the home network 2.

Following GPRS attach by the UE 1, the UE must "discover" the identity (i.e. IP address) of the P-CSCF which it should use. This is done using one of the following mechanisms:

1. Use of DHCP to provide the UE with the domain name of a Proxy-CSCF and the address of a Domain Name Server (DNS) that is capable of resolving the Proxy-CSCF name.

2. Transfer of a Proxy-CSCF address within the PDP Context Activation signalling to the UE (this second alternative is used for terminals not supporting DHCP).

The UE 1 will then notify the S-CSCF 11 of its current location, i.e. the IP address allocated by the GGSN, via the P-CSCF 10 (this process requires authentication of the UE 1 to the S-CSCF and *vice versa* and makes use of the unique subscriber identity). The S-CSCF 11 makes this information available to a Home Subscriber Server 12 which is used to route subsequent incoming calls to the UE 1.

Illustrated in Figure 3 is a UE 13 belonging to a subscriber referred to below as the B-subscriber. The UE 13 is attached to its own home network 14. This network 14 mirrors the home network 2 used by the UE 1, and like numerals, suffixed with a "b", are used to identify components of the network 14. The following discussion assumes that the UE 1 or "A-subscriber" wishes to establish a multimedia call to the UE 13 or "B-subscriber" using the packet switched domain. The UE 1 first sends a SIP INVITE message to the P-CSCF node 10. The INVITE message contains a SIP address of the UE 13 (e.g. john@example.org) as well as an identification of the service required. The P-CSCF node 10 forwards the INVITE message to the S-CSCF node 11.

The S-CSCF 11 verifies the rights of the UE 1 (or rather the subscriber using the UE 1) to use the requested service which is identified in the INVITE message. The S-CSCF 11 must then identify the IP address of the UE 13. It does this by using a look-up table mapping SIP addresses to IP addresses. For a given SIP address, the table provides the IP address of the "home" network of the corresponding subscriber. The identified IP address is used to forward the INVITE message to the S-CSCF 11b in the B-subscriber's home IMS network 9b. Using the SIP address contained in the INVITE message, the S-CSCF 11b identifies the current IP address of the UE 13, and forwards the INVITE message to that address. Upon receipt of the INVITE message, and

assuming that the UE 13 answers the call, an OK message is returned to the UE 1. Typically this message is sent via the two S-CSCFs 11,11b. In order to confirm that the OK message is correctly received by the UE 1, that UE will upon receipt of the message return an ACK message to the peer UE 13. If UE 13 does not receive an ACK message within some predefined time period, it will retransmit the OK message.

As well as its use in establishing PS sessions between mobile terminals, SIP may also be used to establish PS sessions between mobile and fixed terminals and between only fixed terminals. For example, SIP may be used to establish a PS session between a mobile subscriber and a fixed terminal which has a broadband session to the Internet.

As mentioned above, the quality of the packet switched links between the UEs 1,13 and the respective UTRANs may be such that these links are not suitable for transporting real time conversational data, such as voice and video data associated with a call, between the two peer UEs or between one of the UEs and a fixed terminal. Thus, it may be necessary to establish a circuit switched session between the or each UE 1,13 and its circuit switched core network 4,4b. The following mechanism is used to establish these circuit switched sessions.

A UE is assumed to have a PS domain session to the IMS of its home network, and the UE is registered with the IMS domain. The SIP UA of the UE has already informed its IMS SIP server (which will typically be the S-CSCF of the home network, but could be a P-CSCF of a visited network), e.g. during registration, that conversational bearers should not be established over the PS domain, and that the SIP UA will use the CS domain for such bearers (this requirement may be a default setting for the UE). However, the PS domain and the SIP server are used to convey signalling to set up the conversational bearers over the CS domain. It will be understood that, rather than the UE signalling to the SIP server that conversational bearers should be set up over the CS domain, this requirement may already be know to the SIP server (e.g. it could be a "property" defined for the subscriber), or the SIP server may be informed of the requirement by a visited network used by the UE as its access network.

This procedure is illustrated in Figure 4, where a conversational bearer is established between a UE 20 (attached via a UTRAN of a 3G network to PS and CS core networks 21,23) and a fixed terminal 24 (e.g. a PC having a broadband Internet session). A gateway 25 provides a node for terminating CS domain sessions between the UE 20 and the home network. The structure of the networks (CS, PS, IMS) associated with the home operator of the UE 20 are not shown in detail in the Figure, although it will be appreciated that these will take the form illustrated in Figure 3. The gateway 25 communicates with the SIP server 26 (S-CSCF) of the home network IMS 26, e.g. via an IP backbone network (not shown). The SIP server 26 sees the gateway 25 as an application server which will provide a service to the UE 20. The gateway acts, from the SIP point of view, as a transparent Back-to-Back UA (and can modify the Session Description Protocol in a SIP message). Alternatively, the gateway may act as a non-transparent Back-to-Back UA. As the gateway 25 is present not only in the media path, but also in the signalling path, the gateway can provide SIP/SS7 interworking functions. (The SIP server and the gateway may, in some implementations, be physically co-located, either in the home network of a subscriber or in a visited network.)

It is important from the point of view of network operating efficiency that the media gateway (MGW) selected for terminating the circuit switched call is located as close as possible to the RNC of the radio access network. There should be only one physical MGW on the originating side; there should be only one physical MGW on the terminating side. Note that a MGW may be controlled by an MSC and MGCF simultaneously. The MSC, by using the current procedures, shall be entity that selects the MGW. Note that a MGW may be controlled by an MSC and MGCF simultaneously. A single MGCF cannot control all of the MGWs in the operator's network. It is assumed that multiple MGCF will be in operation. Also it should be assumed that for each MSC controlling a set of MGWs there is one determined MGCF controlling the same set of MGWs (i.e., one-to-one MSC-MGCF relation). So the requirement can be rephrased as: There must be a procedure by which the S-CSCF has to be able to select the MGCF associated to the MSC that will be handling the CS call.

Extensions to the CS domain protocols and architecture are not allowed, since the CS domain is already deployed. Extension to the IMS protocols and architecture are allowed, since it is not already deployed.

In order to solve this problem the MGCF does not reserve an IP address and port in the MGW at reception of the SIP INVITE request from the originating UE. Instead, the originating UE inserts a null IP address (0.0.0.0) into the SDP of the INVITE request, and the terminating UE inserts a null IP address (0.0.0.0) into the SDP of the SIP 183 response. The actual IP address where to send media packets is exchanged at a later exchange (SIP UPDATE request), when the Media Gateway has been selected.

The MGCF refers the UE to make the CS call by sending a SIP REFER request. This effectively means that the MSC selects the MGW first. Using BICC protocol it is possible to do a backward bearer setup where the MSC selects the MGW prior to signalling to the MGCF. The MSC includes the MGW identity (BCU-Id) into the BICC IAM message. This allows the MGCF to select the same MGW as the MSC did. The S-CSCF has to know which is the serving MSC of the UE. The serving MSC of the UE is known by the HLR. Since the S-CSCF has an interface to the HSS and the HSS is an evolution of the HLR, the HSS can provide the serving MSC identity. We provide two technical solutions in this step:

a) Pull model: for every SIP INVITE request received at the S-CSCF, the S-CSCF queries the HSS to find out the identity of the serving MSC or the corresponding MGCF address. This solution has the drawback of adding new delays to the call setup and more reduces the capacity of the S-CSCF and HSS with extra queries.

b) Push model: at SIP registration time, the HSS sends to the S-CSCF the serving MSC identity or the corresponding MGCF address. Should there be a change in such serving MSC, the HSS will push the new serving MSC identity or the corresponding MGCF address to the S-CSCF. This is the preferred solution as it does not have any apparent drawback.

With all these procedures in place, the S-CSCF is able to route the SIP INVITE request to the correct MGCF. In addition to this, the IP addresses and ports of the originating side MGW and the terminating side MGW have to be exchanged. This can be done with a SIP UPDATE request sent by both UEs. The SDP in these SIP UPDATE requests will be modified by the MGCFs by changing the null IP address with the actual MGW IP address and port number.

The sequences of Figures 5 and 6 depict the solution in the originating and the terminating mobile network. Note that some irrelevant messages, such us the 200 OK answer for the SIP UPDATE request, are not shown in the sequences.

This procedure creates a one-to-one relation between MSC and MGCF. The HSS informs the S-CSCF of the MSC and/or MGCF handling the CS call. The S-CSCF routes to such MGCF. The MGCF chooses the same MGW as the MSC due to backward bearer setup.

Claims

1. A method of setting up a session between peer nodes of a communication system, said session extending at least in part across a circuit switched access network, the method comprising:

sending a session initiation message from a first node to a second node via a control node over a packet switched access network;

identifying at said control node an appropriate MGCF for the session;

signalling from the control node to the first node an access number associated with the MGCF;

calling said access number from the first node and signalling to the MGCF the identity of the selected MGw;

terminating said call at the selected MGw; and

sending an update message from the first node to the second node via the control node over the packet switched access network, the control node incorporating into the update message an address of said selected MGw.

2. A method according to claim 1, wherein the protocol used to set-up the session is SIP.

3. A method according to claim 1 or 2, wherein said session initiation message is sent via the identified MGCF, and the access number is sent to the first node from the MGCF via the control node.

Abstract

CONVERSATIONAL BEARER NEGOTIATION

A method of setting up a session between peer nodes of a communication system, said session extending at least in part across a circuit switched access network, the method comprising:

sending a session initiation message from a first node to a second node via a control node over a packet switched access network;

identifying at said control node an appropriate MGCF for the session;

signalling from the control node to the first node an access number associated with the MGCF;

calling said access number from the first node and signalling to the MGCF the identity of the selected MGw;

terminating said call at the selected MGw; and

sending an update message from the first node to the second node via the control node over the packet switched access network, the control node incorporating into the update message an address of said selected MGw.

Figure 4

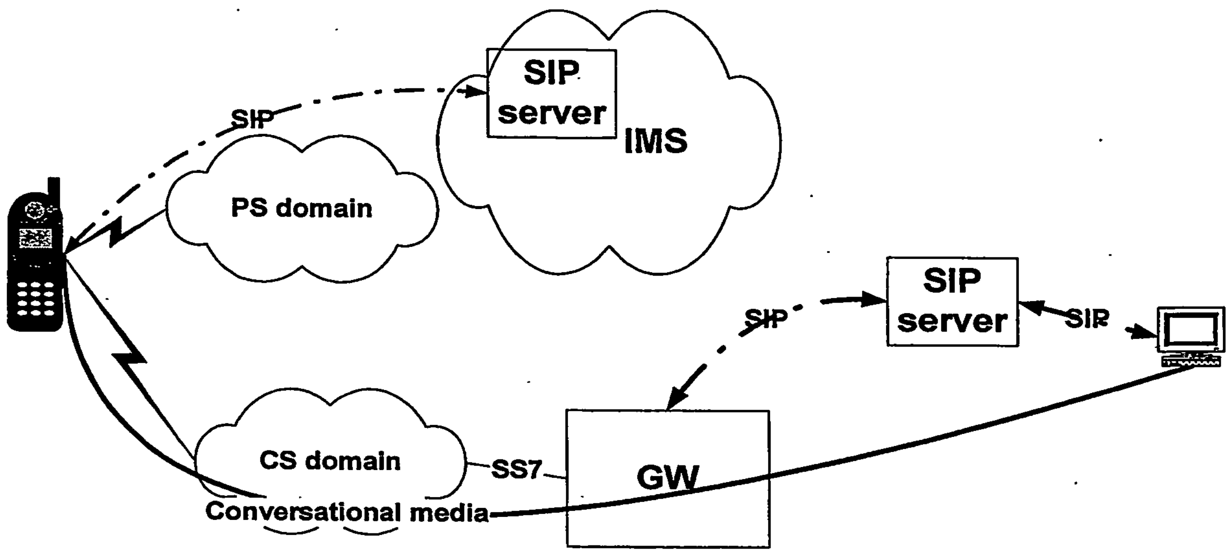

Figure 1

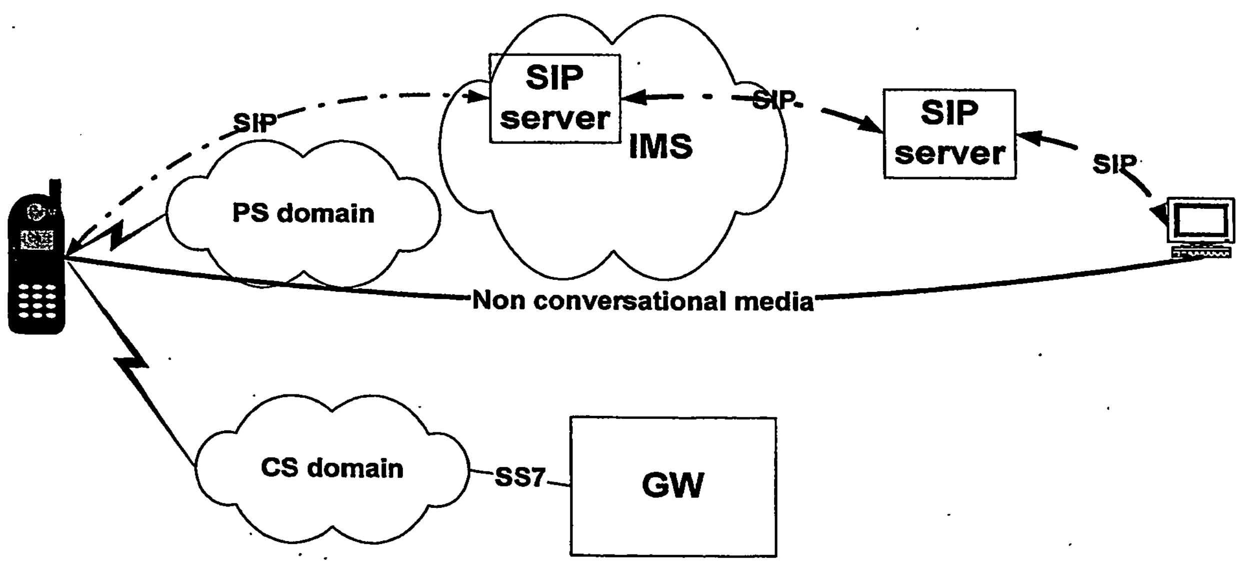

Figure 2

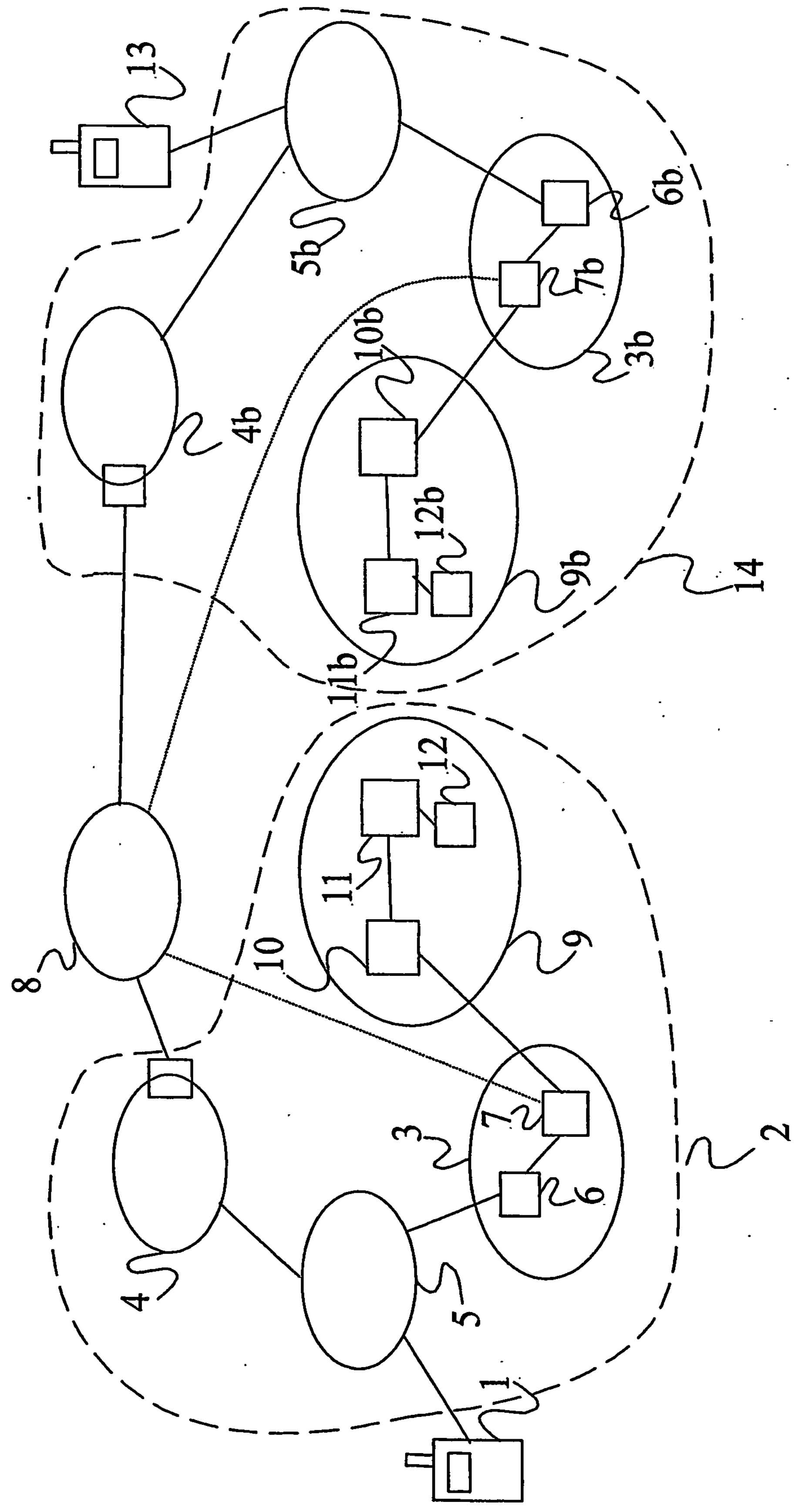

Figure 3

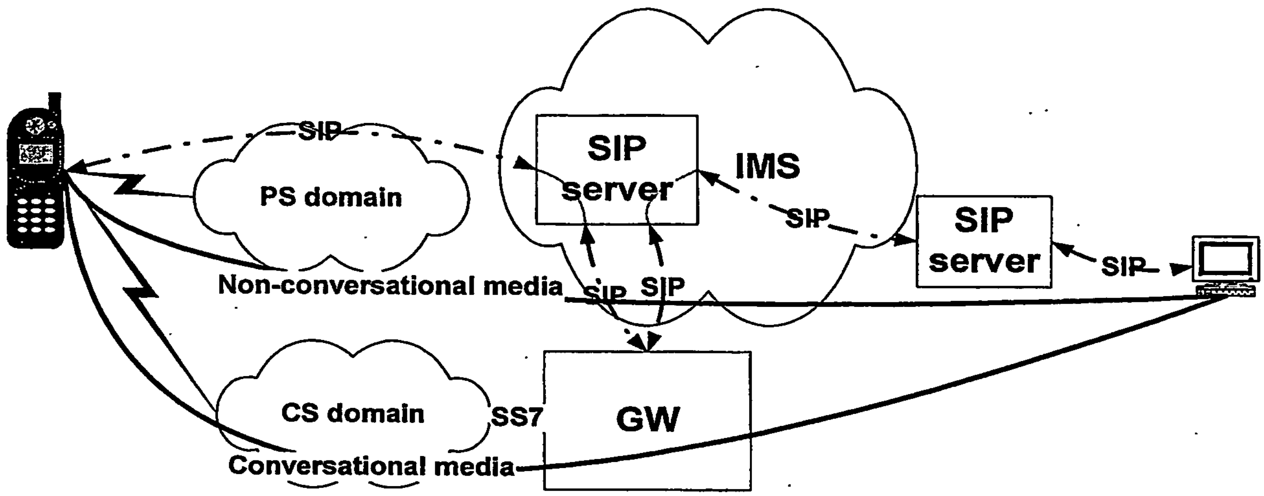
SIP
PS domain
SIP server
IMS
SIP
SIP server
SIP
Non-conversational media
SIP
SIP
CS domain
SS7
GW
Conversational media

Figure 4

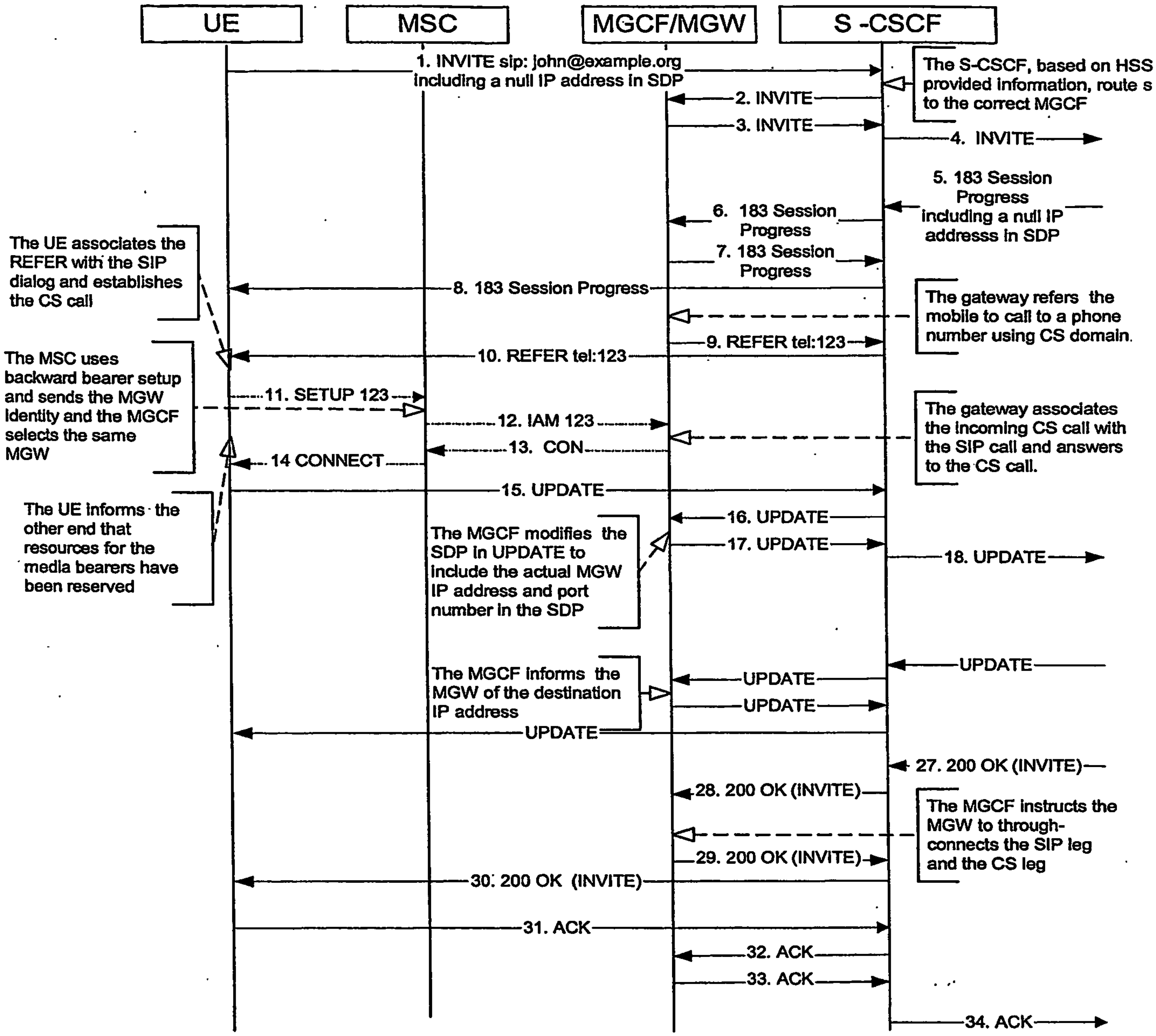
UE
MSC
MGCF/MGW
S -CSCF
1. INVITE sip: john@example.org including a null IP address in SDP
The S-CSCF, based on HSS provided information, route s to the correct MGCF
2. INVITE
3. INVITE
4. INVITE
5. 183 Session Progress including a null IP addresss in SDP
6. 183 Session Progress
7. 183 Session Progress
The UE associates the REFER with the SIP dialog and establishes the CS call
8. 183 Session Progress
The gateway refers the mobile to call to a phone number using CS domain.
9. REFER tel:123
10. REFER tel:123
The MSC uses backward bearer setup and sends the MGW identity and the MGCF selects the same MGW
11. SETUP 123
12. IAM 123
13. CON
14 CONNECT
The gateway associates the incoming CS call with the SIP call and answers to the CS call.
15. UPDATE
16. UPDATE
17. UPDATE
18. UPDATE
The UE informs the other end that resources for the media bearers have been reserved
The MGCF modifies the SDP in UPDATE to include the actual MGW IP address and port number in the SDP
UPDATE
UPDATE
UPDATE
The MGCF informs the MGW of the destination IP address
UPDATE
27. 200 OK (INVITE)
28. 200 OK (INVITE)
The MGCF instructs the MGW to through-connects the SIP leg and the CS leg
29. 200 OK (INVITE)
30. 200 OK (INVITE)
31. ACK
32. ACK
33. ACK
34. ACK

Figure 5

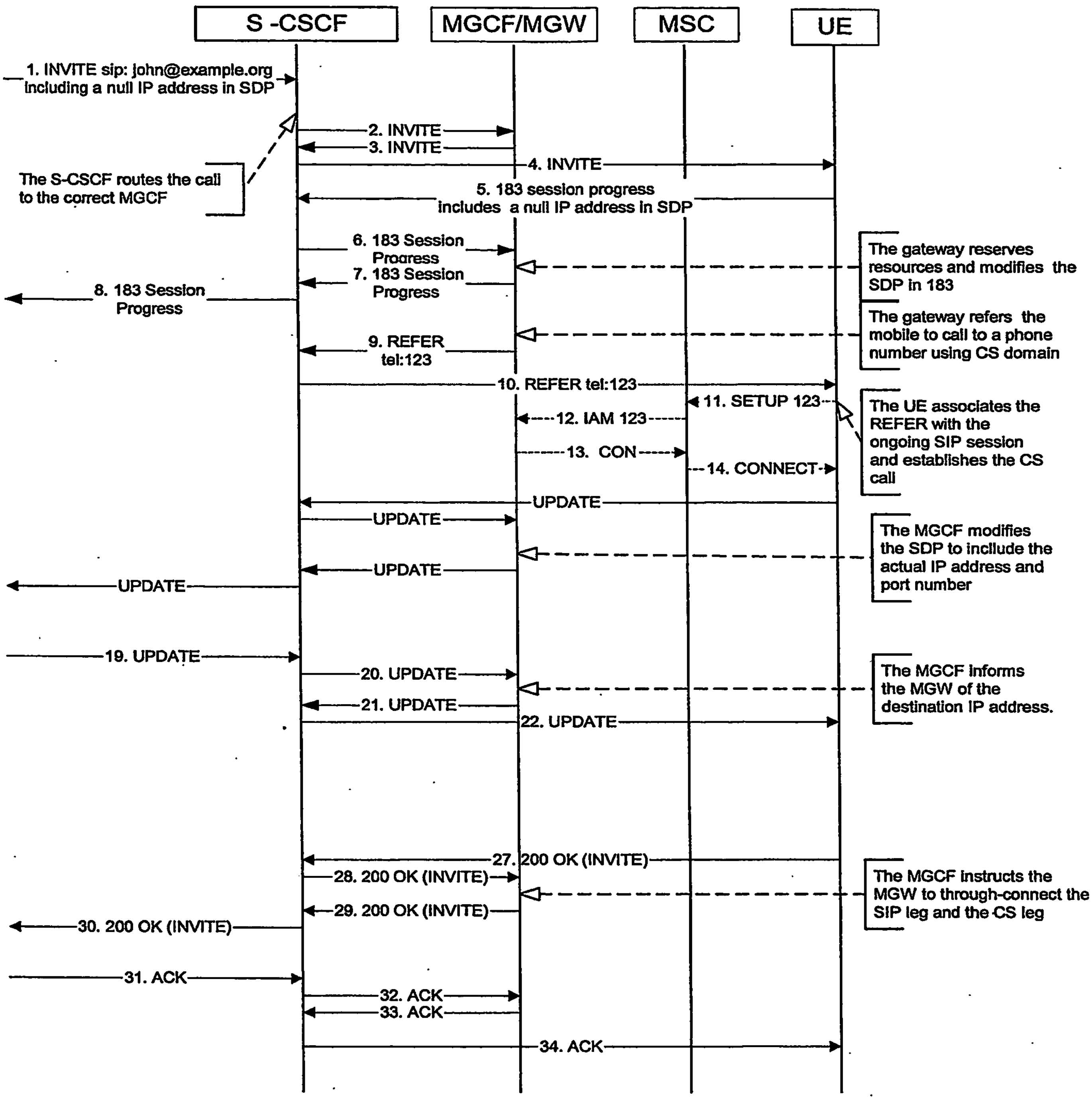
S -CSCF
MGCF/MGW
MSC
UE
1. INVITE sip: john@example.org including a null IP address in SDP
The S-CSCF routes the call to the correct MGCF
2. INVITE
3. INVITE
4. INVITE
5. 183 session progress includes a null IP address in SDP
6. 183 Session Progress
7. 183 Session Progress
The gateway reserves resources and modifies the SDP in 183
8. 183 Session Progress
9. REFER tel:123
The gateway refers the mobile to call to a phone number using CS domain
10. REFER tel:123
11. SETUP 123
12. IAM 123
13. CON
14. CONNECT
The UE associates the REFER with the ongoing SIP session and establishes the CS call
UPDATE
UPDATE
UPDATE
UPDATE
The MGCF modifies the SDP to include the actual IP address and port number
19. UPDATE
20. UPDATE
21. UPDATE
22. UPDATE
The MGCF informs the MGW of the destination IP address.
27. 200 OK (INVITE)
28. 200 OK (INVITE)
29. 200 OK (INVITE)
30. 200 OK (INVITE)
The MGCF instructs the MGW to through-connect the SIP leg and the CS leg
31. ACK
32. ACK
33. ACK
34. ACK

Figure 6